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EXAMINER	
GODBOLD, DOUGLAS	

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Please find below and/or attached an Office communication concerning this application or proceeding.

The time period for reply, if any, is set in the attached communication.

Office Action Summary	Application No. 10/695,125	Applicant(s) SINGHAL, MANOJ	
	Examiner DOUGLAS C. GODBOLD	**Art Unit** 2626	

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE _3_ MONTH(S) OR THIRTY (30) DAYS, WHICHEVER IS LONGER, FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☒ Responsive to communication(s) filed on _25 February 2008_.

2a)☐ This action is **FINAL**. 2b)☒ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) _1-6 and 8-26_ is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) _1-6 and 8-26_ is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☐ The specification is objected to by the Examiner.

10)☒ The drawing(s) filed on _25 February 2008_ is/are: a)☒ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

Priority under 35 U.S.C. § 119

12)☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☐ All b)☐ Some * c)☐ None of:

1.☐ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☒ Notice of References Cited (PTO-892)
2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)
3) ☐ Information Disclosure Statement(s) (PTO/SB/08) Paper No(s)/Mail Date _____.
4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____ .
5) ☐ Notice of Informal Patent Application
6) ☐ Other: _____.

DETAILED ACTION

1. This Office Action is in response to correspondence filed February 25, 2008 in references to application 10/695,125.

Continued Examination Under 37 CFR 1.114

2. A request for continued examination under 37 CFR 1.114, including the fee set forth in 37 CFR 1.17(e), was filed in this application after final rejection. Since this application is eligible for continued examination under 37 CFR 1.114, and the fee set forth in 37 CFR 1.17(e) has been timely paid, the finality of the previous Office action has been withdrawn pursuant to 37 CFR 1.114. Applicant's submission filed on February 25, 2008 has been entered.

Response to Amendment

3. The Amendment filed February 25, 2008 has been accepted and considered in this office action. The Drawings have been amended, and the objection to the drawings has been withdrawn. Claims 1, 5, 6, 16, and 18 have been amended, and claims 25 and 26 added. The rejections of claims 5 and 6 under 35 U.S.C. 112 have been withdrawn.

Response to Arguments

4. Applicant's arguments filed February 25, 2008 have been fully considered but they are not persuasive.

5. In response to applicant's argument that the references fail to show certain features of applicant’s invention, it is noted that the features upon which applicant relies (i.e., Remarks page 10, that audio components are frequency components and not time components) are not recited in the rejected claim(s). Although the claims are interpreted in light of the specification, limitations from the specification are not read into the claims. See *In re Van Geuns*, 988 F.2d 1181, 26 USPQ2d 1057 (Fed. Cir. 1993).

6. In response to applicant's argument, see remarks page 13, that the filter of Pohlmann does not reduce audio data, the examiner respectfully disagrees. Although the purpose of the filter is to prevent aliasing, all audio data above the filter is deleted, and therefore audio data is reduced.

7. In response to applicant's argument, see remarks page 13, Polhmann does not teach the decimator of claim 18, the examiner respectfully disagrees. The cited passage of Pohlmann discloses removing one sample every 23ms, which fits the definition of a decimator, where N= the number of frames in 23ms. Although the purposes discussed in Pohlmann is to couple audio devices by reducing the data rate, one of ordinary skill in the art can appreciate the benefits of reducing the data rate for other applications.

Claim Rejections - 35 USC § 112

8. The following is a quotation of the first paragraph of 35 U.S.C. 112:

> The specification shall contain a written description of the invention, and of the manner and process of making and using it, in such full, clear, concise, and exact terms as to enable any person skilled in the art to which it pertains, or with which it is most nearly connected, to make and use the same and shall set forth the best mode contemplated by the inventor of carrying out his invention.

9. Claims 25 and 26 rejected under 35 U.S.C. 112, first paragraph, as failing to comply with the written description requirement. The claim(s) contains subject matter which was not described in the specification in such a way as to reasonably convey to one skilled in the relevant art that the inventor(s), at the time the application was filed, had possession of the claimed invention. The specification makes no mention that the selected audio signal components are frequency components. Only windowing is used for classification analysis. The only mention of frequency domain signals is in paragraphs 0072-0079, and these are only used for encoding a signal for transmission, NOT the analysis of the signal itself for classification. The claimed invention is based on zero-crossing rates. It would be an impossibility to do this in the frequency domain.

10. The following is a quotation of the second paragraph of 35 U.S.C. 112:

> The specification shall conclude with one or more claims particularly pointing out and distinctly claiming the subject matter which the applicant regards as his invention.

11. Claim 19 rejected under 35 U.S.C. 112, second paragraph, as being indefinite for failing to particularly point out and distinctly claim the subject matter which applicant regards as the invention.

12. Claim 19 modifies a limitation that is not required by claim 18. Claim 18 recited "one or both of" a low-pass filter and a decimator. Claim 19 is directed toward the

decimator, which is not a required limitation. If only the low-pass filter is selected, then claim 19 has no antecedent basis.

Claim Rejections - 35 USC § 102

13. The following is a quotation of the appropriate paragraphs of 35 U.S.C. 102 that form the basis for the rejections under this section made in this Office action:

> A person shall be entitled to a patent unless –
>
> (b) the invention was patented or described in a printed publication in this or a foreign country or in public use or on sale in this country, more than one year prior to the date of application for patent in the United States.

14. Claims 1, 3, 4, 10, 11, 13, and 14 rejected under 35 U.S.C. 102(b) as being anticipated by Saunders (Real-Time Discrimination of Broadcast Speech/Music).

15. Consider claim 1, Saunders teaches a method for classifying an audio signal (we describe a technique which is successful at discriminating speech from music; page 993, column 1, line 1), the method comprising:

receiving an audio signal to be classified (this is a technique for discriminating speech from music from an FM broadcast; page 993, column 1, line 2);

analyzing selected audio signal components (The first step is to measure the ZCR of the signal over a 2.4 second segment of the data; page 994, column 2, line 43);

recording a result of analysis of the selected audio signal components (would be inherent in order to compare it);

comparing the recorded result of analysis to a threshold value (If this statistic exceeds a specific threshold, the distribution outside these bounds is significantly skewed and the waveform is likely speech; page 994, column 2, line 43); and

classifying the audio signal based upon comparison of the recorded result of analysis and the threshold value (If this statistic exceeds a specific threshold, the distribution outside these bounds is significantly skewed and the waveform is likely speech; page 994, column 2, line 43).

16. Consider claim 3, Saunders teaches the method according to claim 1, wherein analyzing the selected audio signal components comprises counting zero point transitions of the selected audio signal components (The first step is to measure the ZCR of the signal over a 2.4 second segment of the data; page 994, column 2, line 43. Measuring the Zero Crossing Rate would entail counting the number of zero crossings).

17. Consider claim 4, Saunders teaches the method according to claim 1, wherein recording a result of analysis of the selected audio signal components comprises recording a count value of a number of zero point transitions of the selected audio signal components (The first step is to measure the ZCR of the signal over a 2.4 second segment of the data; page 994, column 2, line 43. Measuring the Zero Crossing Rate would entail counting the number of zero crossings. This number would inherently have to be stored somewhere in order to process it or manipulate it).

18. Consider claim 10, Saunders teaches the method according to claim 1, wherein classifying the audio signal occurs at a receiving end of an audio transmission system (this is a technique for discriminating speech from music from an FM broadcast; page 993, column 1, line 2).

19. Consider claim 11, Saunders teaches the method according to claim 1, wherein the audio signal is one of an analog signal and a digital signal (A sample rate of 16Khz was chosen for this discrimination technique; page 995, column 1 line 1. If something is sampled it is well understood that it is being converted to a digital signal. this is a technique for discriminating speech from music from an FM broadcast; page 993, column 1, line 2. This further tells us that the signal started out as an analog signal as at the time of the publication of Saunders all FM broadcasts were analog.).

20. Consider claim 13, Saunders teaches the method according to claim 1, wherein the threshold value used in the comparison determined through trial and error of a plurality of iterations in a comparing device (Data was collected manually by listening, collecting and storing features, and labeling the segment. A variety of content was processed, including talk, commercials, and many types of music. Once the classifier was trained, the parameters were stored and fed into the real-time feature extraction/classifier routine; page 995, column 1, line 33).

21. Consider claim 14, Saunders teaches the method according to claim 1, wherein analyzing selected audio signal components comprises counting zero point transitions of the audio signal for a predetermined period of time (The first step is to measure the ZCR of the signal over a 2.4 second segment of the data; page 994, column 2, line 43. Measuring the Zero Crossing Rate would entail counting the number of zero crossings).

Claim Rejections - 35 USC § 103

22. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

23. Claim 2 is rejected under 35 U.S.C. 103(a) as being unpatentable over Saunders in view of Carey (A Comparison of Features for Speech, Music Discrimination).

24. Consider claim 2, Saunders in view of Benyassine teaches the method according to claim 1, but does not specifically teach wherein classifying the audio signal based upon comparison of the recorded result of analysis and the threshold value further comprises:

if the recorded result of analysis is greater than the threshold value, then the audio signal is determined to be music; and

if the recorded result of analysis is less than the threshold value, then the audio signal is determined to be speech.

In the same field of speech/music discrimination, Carey teaches if the recorded result of analysis is greater than the threshold value, then the audio signal is determined to be music (table 1 shows that the mean value of number of zero crossing (u) for music 0.18 is greater than that of speech 0.17); and

if the recorded result of analysis is less than the threshold value, then the audio signal is determined to be speech (table 1 shows the mean value of zero crossing for speech 0.17 was less that music 0.18).

Although Saunders in view of Benyassine uses a slightly different zero crossing analysis method than does Carey, it would have been obvious to one of ordinary skill in the art at the time of the invention to use the parameters of Carey as this method would be computationally inexpensive (Carey page 151, column 2, section 4.4).

25. Claims 5, 8, 16, and 18-24 are rejected under 35 U.S.C. 103(a) as being unpatentable over Saunders in view of Benyassine and further in view of Pohlmann (Principles of Digital Audio).

26. Consider claim 5, Saunders teaches the method according to claim 1, but does not specifically teach further comprising selecting audio signal components prior to analyzing selected audio signal components (The first step is to measure the ZCR of the signal over a 2.4 second segment of the data; page 994, column 2, line 43. Inherent

the segment is selected first), wherein said selecting audio signal components comprises passing the audio signal through a low pass filter for filtering out audio signal components having a frequency greater than a predetermined frequency thereby reducing an amount of audio information to be analyzed

In the same field of audio analysis, Benyassine teaches transmitting an audio signal using encoder 112 of figure 1 that samples at a rate of 8000Hz; column 3, line 60.

Therefore it would have been obvious to combine the sampling of audio for transmitting of Benyassine with the classification system of Saunders in order to allow the transmissions of digital signals.

This does not teach specifically wherein said selecting audio signal components comprises passing the audio signal through a low pass filter for filtering out audio signal components having a frequency greater than a predetermined frequency thereby reducing an amount of audio information to be analyzed.

In the same field of audio encoding, Pohlmann suggests passing the audio signal through a low pass filter for filtering out audio signal components having a frequency greater than a predetermined frequency thereby reducing an amount of audio information to be analyzed (sampled audio must be passed through a low pass filter at the Nyquist frequency in order to prevent distortion called aliasing; page 30, prevention section. By Nyquist filtering, signal components have been reduced.).

Therefore it would have been obvious to combine the sampling of Benyassine with the filtering of Pohlmann in order to prevent aliasing, and to provide a way to digitize the audio signal for analysis, coding and transmission.

27. Consider claim 8, Saunders teaches the method according to claim 1, further comprising:

selecting a number of transmitted audio signal components for analysis (The first step is to measure the ZCR of the signal over a 2.4 second segment of the data; page 994, column 2, line 43.).

However Saunders does not specifically teach transmitting components of the audio signal having a frequency less than a predetermined frequency.

In the same field of audio analysis, Benyassine teaches transmitting an audio signal using encoder 112 of figure 1 that samples at a rate of 8000Hz; column 3, line 60.

Therefore it would have been obvious to combine the sampling of audio for transmitting of Benyassine with the classification system of Saunders in order to allow the transmissions of digital signals.

This does not say specifically that that the audio being transmitted is less than a predetermined frequency.

In the same field of audio encoding, Pohlmann teaches that sampled audio must be passed through a low pass filter at the Nyquist frequency in order to prevent distortion called aliasing; page 30, prevention section.

Therefore it would have been obvious to combine the sampling of Benyassine with the filtering of Pohlmann in order to prevent aliasing, and to provide a way to digitize the audio signal for analysis, coding and transmission.

28. Consider claim 16, Saunders teaches an apparatus for classifying an audio signal (The experimental setup used a Gradient A/D unit attached to a workstation; page 995, column 1, line 38), the apparatus comprising:

a zero point counter for counting and recording zero point transitions encountered in analysis of the selected audio signal components (The first step is to measure the ZCR of the signal over a 2.4 second segment of the data; page 994, column 2, line 43); and

a comparator for comparing a recorded result of analysis to a threshold value and classifying the audio signal based upon comparison of the recorded result of analysis and the threshold value (If this statistic exceeds a specific threshold, the distribution outside these bounds is significantly skewed and the waveform is likely speech; page 994, column 2, line 43).

However Saunders does not specifically teach a circuit for packetizing the audio signal into packets, said packets including a header, said header including a flag indicating classification of the audio signal.

In the same field of music and speech discrimination Benyassine teaches a circuit for packetizing the audio signal into packets (The encoder 112 segments the digitized speech signal into frames to generate a bitstream. In one embodiment, the speech coding system 100 uses frames having 160 samples and corresponding to 20 milliseconds per frame at a sampling rate of about 8000 Hz. The encoder 112 provides the frames via a bitstream to the communication medium 104; column 3, line 56. A window is the same as a packet in this use in the communication medium.

communication devices 102 and 106 may be cellular telephones radios, or VoIP systems; column 3 line 6-11. Cell phones and VoIP systems both used packetized data), said packets including a header, said header including a flag indicating classification of the audio signal (all flags used to mark audio frames are shown in Table 1, column 9. The music detection flag F_M is set if either threshold for music conditions are met; column 7 line 37).

This combination of Saunders and Benyassine does not specifically teach at least one audio signal component reducer for selecting a reduced number of audio signal components for analysis.

In the same field of audio processing, Pohlmann teaches, at least one audio signal component reducer for selecting a reduced number of audio signal components for analysis (This is nothing more than resampling the audio signal As noted many different sampling rates are used, devices cannot be connected when their sampling rates differ… For example a 44.1kHz signal can be converted to 44.056kHz by removing one sample every 23ms; page 460, fist full paragraph This could be carried to the extreme of reducing the sampling rate more drastically, such as converting from 44kHz to 22kHz by dropping every other sample).

Therefore it would have been obvious to one of ordinary skill in the art at the time of the invention to resample the signal as taught by Pohlmann in the system of Saunders in order to reduce the data rate in the system and thereby lowering processing requirements.

29. Consider claim 18, Pohlmann teaches the apparatus according to claim 16, wherein the at least one audio signal component reducer comprises one or both of: a low pass filter that prevents transmission of components of the audio signal having a frequency greater than a predetermined frequency; and a decimator (This is nothing more than resampling the audio signal As noted many different sampling rates are used, devices cannot be connected when their sampling rates differ… For example a 44.1kHz signal can be converted to 44.056kHz by removing one sample every 23ms; page 460, fist full paragraph This could be carried to the extreme of reducing the sampling rate more drastically, such as converting from 44kHz to 22kHz by dropping every other sample)

30. Consider claim 19, Pohlmann teaches the apparatus according to claim 18, wherein the decimator selecting a reduced number of audio components for analysis comprises the decimator selecting every 1 in N audio signal components to be transmitted and selecting the audio signal components between 1 and N to be discarded (This is nothing more than resampling the audio signal As noted many different sampling rates are used, devices cannot be connected when their sampling rates differ… For example a 44.1kHz signal can be converted to 44.056kHz by removing one sample every 23ms; page 460, fist full paragraph. This could be carried to the extreme of reducing the sampling rate more drastically, such as converting from 44kHz to 22kHz by dropping every other sample).

31. Consider claim 20, Benyassine teaches the apparatus according to claim 16, further comprising at least one of an audio signal encoder (figure 1, encoder 112) and an audio signal decoder (figure 1, decoder 114).

32. Consider claim 21, Benyassine teaches the apparatus according to claim 20, further comprising a speech/music classifying device being associated with the audio signal encoder (Figure 1, encoder 112, part of transmission side, may contain a music classifier with voice activity detector; column 3, line 62.).

33. Consider claim 22, Saunders teaches the apparatus according to claim 20, further comprising a speech/music classifying device being associated with the audio signal decoder (this is a technique for discriminating speech from music from an FM broadcast; page 993, column 1, line 2. An FM signal must be decoded before it can be classified or played or manipulated in anyway).

34. Consider claim 23, Saunders teaches the apparatus according to claim 20, further comprising a signal processor and an audio processing unit associated with the audio signal decoder (The experimental setup used a Gradient A/D unit attached to a workstation; page 995, column 1, line 38. Using data processed on the fly and tuning the radio dial at will, the classification accuracy averaged between 95 and 96%; page 995, column 1, line 43. This is a technique for discriminating speech from music from

an FM broadcast; page 993, column 1, line 2. An FM signal must be decoded before it can be classified or played or manipulated in anyway).

35. Consider claim 24, Benyassine teaches the apparatus according to claim 20, further comprising a bitstream multiplexer associated with the audio signal decoder (signal (communication devices 102 and 106 may be cellular telephones radios, or VoIP systems; column 3 line 6-11. Cell phones and VoIP systems both used packetized data. It is inherent that some kind of multiplexing must be employed in order to packetize the data).

36. Claim 6 is rejected under 35 U.S.C. 103(a) as being unpatentable over Saunders in view of Pohlmann.

37. Consider claim 6, Saunders teaches the method according to claim 1, further comprising selecting audio signal components prior to analyzing selected audio signal components (The first step is to measure the ZCR of the signal over a 2.4 second segment of the data; page 994, column 2, line 43. Inherent the segment is selected first), but does not specifically teach wherein said selecting audio signal components comprises passing the audio signal through a decimator, wherein every 1 in N audio signal components is transmitted and audio signal components between 1 and N are discarded.

In the same field of audio processing, Pohlmann teaches, wherein said selecting audio signal components comprises passing the audio signal through a decimator, wherein every 1 in N audio signal components is transmitted and audio signal components between 1 and N are discarded (This is nothing more than resampling the audio signal As noted many different sampling rates are used, devices cannot be connected when their sampling rates differ… For example a 44.1kHz signal can be converted to 44.056kHz by removing one sample every 23ms; page 460, fist full paragraph This could be carried to the extreme of reducing the sampling rate more drastically, such as converting from 44kHz to 22kHz by dropping every other sample).

Therefore it would have been obvious to one of ordinary skill in the art at the time of the invention to resample the signal as taught by Pohlmann in the system of Saunders in order to reduce the data rate in the system and thereby lowering processing requirements.

38. Claims 9, 15 are rejected under 35 U.S.C. 103(a) as being unpatentable over Saunders in view of Benyassine.

39. Consider claim 9, Saunders teaches the method according to claim 1, but does not teach specifically wherein classifying the audio signal occurs at a transmitting end of an audio transmission system.

However in the same field of music and speech discrimination Benyassine teaches classifying the audio signals at a transmitting end of an audio transmission

system (Figure 1, encoder 112, part of transmission side, may contain a music classifier with voice activity detector; column 3, line 62.)

Therefore it would have been obvious to one of ordinary skill in the art at the time of the invention to classify the music or voice at the transmitting side of the system as taught by Benyassine in order to determine properties of the signal in order to best encode the signal for transmission (Benyassine; column 1 line 62 - column 2 line 13).

40. Consider claim 15, Saunders teaches the method according to claim 1, but does not specifically teach further comprising:

converting the audio signal from an analog signal to a digital signal;

encoding the audio signal;

packetizing the audio signal;

transmitting the audio signal;

decoding the audio signal; and

processing the audio signal, wherein processing at least comprises one of storing the audio signal and playing the audio signal.

However in the same field of music and speech discrimination Benyassine teaches converting the audio signal from an analog signal to a digital signal (figure 1, A/D converter 108);

encoding the audio signal (figure 1, encoder 112);

packetizing the audio signal (communication devices 102 and 106 may be cellular telephones radios, or VoIP systems; column 3 line 6-11. Cell phones and VoIP systems both used packetized data);

transmitting the audio signal (figure 1, signals are transmitted over communication medium 104);

decoding the audio signal (using decoder 114, figure 1); and

processing the audio signal, wherein processing at least comprises one of storing the audio signal and playing the audio signal (output of system is synthesized speech signal 120, figure 1).

Therefore it would have been obvious to one of ordinary skill in the art at the time of the invention to use the transmission scheme of Benyassine with the audio classification method of Saunders in order to provide an efficient way to effectively transmit audio signals (Benyassine; column 1 line 62 - column 2 line 13),

41. Claim 12 is rejected under 35 U.S.C. 103(a) as being unpatentable over Saunders. Saunders teaches the method according to claim 1, but does not specifically teach wherein the threshold value used in the comparison is pre-determined and pre-set by a user.

However Saunders does teach Data was collected manually by listening, collecting and storing features, and labeling the segment. A variety of content was processed, including talk, commercials, and many types of music. Once the classifier

was trained, the parameters were stored and fed into the real-time feature extraction/classifier routine; page 995, column 1, line 33.

With data being collected manually, it must be entered manually, and although is not specifically the threshold, one of ordinary skill in the art that the training of the classifier by manually collecting data is changing the threshold. Therefore in fact, the user is in a way changing the threshold value is preset and determined by the user.

42. Claim 17 rejected under 35 U.S.C. 103(a) as being unpatentable over Saunders in v as applied to claim Saunders, Benyassine, and Pohlmann above, and further in view of Carey.

43. Consider claim 17, Saunders, Benyassine, and Pohlmann teaches the apparatus according to claim 16, but does not specifically teach wherein classifying the audio signal based upon comparison of the recorded result of analysis and the threshold value in the comparator further comprises:

if the recorded result of analysis is greater than the threshold value, then the audio signal is determined to be music; and

if the recorded result of analysis is less than the threshold value, then the audio signal is determined to be speech.

In the same field of speech/music discrimination, Carey teaches if the recorded result of analysis is greater than the threshold value, then the audio signal is determined

to be music (table 1 shows that the mean value of number of zero crossing (u) for music 0.18 is greater than that of speech 0.17); and

if the recorded result of analysis is less than the threshold value, then the audio signal is determined to be speech (table 1 shows the mean value of zero crossing for speech 0.17 was less that music 0.18).

Although Saunders, Benyassine, and Pohlmann uses a slightly different zero crossing analysis method than does Carey, it would have been obvious to one of ordinary skill in the art at the time of the invention to use the parameters of Carey as this method would be computationally inexpensive (Carey page 151, column 2, section 4.4).

44. Claim 25 is rejected under 35 U.S.C. 103(a) as being unpatentable over Saunders in view of Benyassine and further in view of Pohlmann as applied to claim 16 above, and further in view of Nishiguchi et al. (US Patent 5,809,455).

Saunders teaches the apparatus according to claim 16, but does not specifically teach wherein the selected audio signal components are frequency components.

In the same field of sound classification, Nishiguchi teaches the selected audio signal components are frequency components (figure 4, signals are subjected through FFT 33 before a decision is made.)

Therefore it would have been obvious to one of ordinary skill in the art to use an FFT to convert the windows to frequency domain in order to allow for other methods like energy distribution to be used to classify the signals.

45. Claim 26 is rejected under 35 U.S.C. 103(a) as being unpatentable over Saunders in view of Nishiguchi et al).

Saunders teaches the method according to claim 1, but does not specifically teach wherein the selected audio signal components are frequency components.

In the same field of sound classification, Nishiguchi teaches the selected audio signal components are frequency components (figure 4, signals are subjected through FFT 33 before a decision is made.)

Therefore it would have been obvious to one of ordinary skill in the art to use an FFT to convert the windows to frequency domain in order to allow for other methods like energy distribution to be used to classify the signals.

Conclusion

Any inquiry concerning this communication or earlier communications from the examiner should be directed to DOUGLAS C. GODBOLD whose telephone number is (571)270-1451. The examiner can normally be reached on Monday-Thursday 7:00am-4:30pm Friday 7:00am-3:30pm.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Patrick Edouard can be reached on (571) 272-7603. The fax phone number for the organization where this application or proceeding is assigned is 571-273-8300.

Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Status information for published applications may be obtained from either Private PAIR or Public PAIR. Status information for unpublished applications is available through Private PAIR only. For more information about the PAIR system, see http://pair-direct.uspto.gov. Should you have questions on access to the Private PAIR system, contact the Electronic Business Center (EBC) at 866-217-9197 (toll-free). If you would like assistance from a USPTO Customer Service Representative or access to the automated information system, call 800-786-9199 (IN USA OR CANADA) or 571-272-1000.

DCG

/Patrick N. Edouard/
Supervisory Patent Examiner, Art Unit 2626